

SIP-Phone User Manual



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Revision History

Version	Date	Author	Software Version	Note
LP_SIP_100	2004/03	Sabrina	For test version only.	First released for software test version.
LP_SIP_100a	2004/04/07	Sabrina	lp101/201sip.100	For first formal released software.
LP_SIP_101	2004/12/1	Sabrina	lp201sip.101	Modify: 1. Forward function 2. Call list function 3. LCD menu firmware upgrading section. 4. Add DDNS function 5. Add VLAN function
LP_SIP_102	2005/3/10	Sabrina	lp201sip.102	Modify: 1. PPPoE Echo 2. EMS 3. SIP-Info DTMF 4. 2 level access

Chapter 1 Overview of the SIP-Phone

SIP-Phone is a full-featured IP-based telephone set via Ethernet base communication. Over the office LAN connection, it provides IP-PBX solution such as station-to-station call, IP call and local PSTN/PBX Extension call via PSTN Gateway.

SIP-Phone provides two 10/100BaseT switch/hub RJ-45 ports allow connecting to office LAN and PC simultaneously. It is compatible with ADSL or Cable Modem provided by ITSP, ISP or Carrier Company to provide VoIP services to residential and SOHO application. SIP-Phone is also an integrated Analog Phone provides IP call or PSTN call selection. When external power is down, it can be a Plain Old Telephone set (POTs).

It provides internal high-quality speakerphone, programmable keys and feature buttons. SIP-Phone also embedded with a dot matrix of two lines 24 characters LCD, which can display date and time, calling party name, calling party number, and digits dialed and etc.

1. Hardware Overview

1. Front View and Keypad function



- ◆ **LCD:** 2 lines, 24 character Dot Matrix display.
- ◆ **C:** Jump out current LCD menu.
- ◆ **←** : Move to previous selection or clear previous data.
- ◆ **→** : Move to right or next selection.
- ◆ **OK:** Press OK to confirm the modification.
- ◆ **Direct Line (DL) Button 1 – 10:** User press DL button after off-hook to do speed dial according to phone book data from 1-10 (please refer to LCD configuration-**3.Phone Book**; Advanced Configurations via Telnet- **10.[pbook] command**, or Web Configuration-Phone Book chapter.

Note:

1. When SIP-Phone is in PSTN mode, only **PSTN** and **SPEAKER** function key can work.
2. On LCD will display "...Incoming Call..." to inform user when SIP-Phone has both IP and PSTN side incoming calls.
3. If in communication with IP side, user can press HOLD to hold IP side, then press PSTN to pick up PSTN side, after that can press HOLD again to retrieve IP side.
4. If in communication with PSTN side, user must hang up PSTN side before pick up IP side.

- ◆ **HOLD:** To hold a call, after press HOLD button, both sides will hear hold tone.
- ◆ **SPEED:**
 1. Press SPEED and number (Phone book index) after off-hook can do speed dial according to phone book data (please refer to 3.LCD configuration-6. **Phone Book** or Advanced Configurations via 4.Telnet- **10. [pbook] command**).
 2. Switch input mode between character mode or digit mode, e.g., when user wants to input phone number can press **SPEED** to switch input mode as digit mode; when user wants to input name can also press **SPEED** to switch input mode as character mode.
- ◆ **FORWARD:** Forward an incoming call to another IP device. (Please refer to *LCD configuration-Forward Type*)
- ◆ **MESSAGE and its indicated LED light:** When having missed incoming calls, the MESSAGE LED will be flashing. User can check the information of missed calls by pressing the MESSAGE button.
- ◆ **TRANSFER:**
 1. Transfer a call to the third site. When A and B are in communication, A wants to transfer this call to C, A can press **TRANSFER** button, now B will hear hold tone, and A will hear dial tone, then A can press phone number of C, after C picks up, A can talk with C, after A hangs up, B and C can be connected.

Note:

1. A cannot press phone number of C before **hearing dial tone**.
2. If A press TRANSFER and number, press TRANSFER again can cancel transferring and retrieve call with B.
3. If A press TRANSFER and number of C, C is ringing, press TRANSFER again C will return to standby mode and A can retrieve call with B.
4. Before C picks up, A cannot hangs up the phone.

2. Change characters to be capital or lowercase: when pressing TRANSFER before press letters can switch input type of letters.

- ◆ **REDIAL:** Redial the last outgoing call.
- ◆ **+ And -:** Adjust the voice volume heard of communication.
- ◆ **SPEAKER:** Hand free mode. User can talk without picking up handset.

Note:

1. All function keys mentioned above (except dialing keypad) are effective only in IP Phone mode.
2. When SIP Phone fail to register to Proxy server under Proxy mode, when user wants to dial out, SIP Phone will play busy tone, and on LCD will display "Register Failed."

2.Back View



- ◆ **DC 9V:** DC 9V power input outlet
- ◆ **LAN:** RJ-45 connector, connected directly to the **Hub** through the **straight** CAT-5 cable.
- ◆ **PC:** RJ-45 connector, connected directly to the **PC** through the **straight** CAT-5 cable
- ◆ **Line:** RJ-11 connector, connected directly to the PSTN analog line.

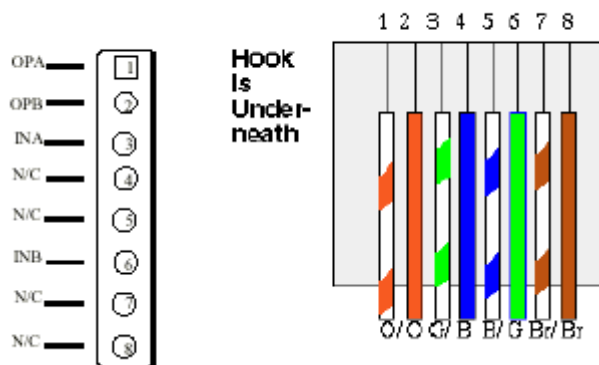
Note:

There are two LED indicated lights: LINK/ACT and 10/100 for LAN port and PC port. When network status is regular, LED of LINK/ACT will light on; when SIP-Phone is transmitting or receiving data, LED will be flashing; when transmit rate is in 10 mbps or 100mbps, LED of 10/100 will light off or light on.

3.Specification of connector

1、 Ethernet Port :

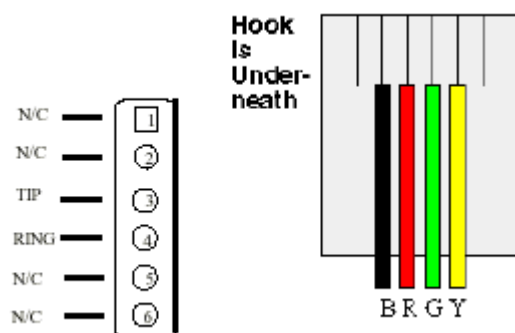
Ethernet port is for connecting SIP-Phone to network, transmit rate supports 10/100 Base-T.



Ethernet connector (LAN)

2、 RJ-11connector :

RJ-11 connector is for connecting SIP-Phone with PSTN.



RJ11connector

2. Software Features and Specification

Application:

- ISP/ITSP (Internet Telephony Service Provider)
- IP-PBX with office telephony services
- Multi-nation enterprise communication
- SOHO Telephony

Calling Features

- Call Hold
- Call Transfer
- Call Forward
- 10 sets last number Redial
- Ten configurable speed dials

Network Supported

- Fixed IP
- Dynamic Host Configuration Protocol (DHCP)
- PPPoE connection (When PPPoE disconnect, SIP-Phone can automatically re-connect)
- Behind NAT IP Sharing Device
- Support QOS by setting DSCP (Differentiated Service Code Point) parameters of VoIP packet
- Support 802.1p1q (VLAN)

Audio Features

- G.711 a/μ-Law, G.723.1, G.729, G.729a
- VAD, CNG
- G.165/G.168 compliant echo cancellation
- Programmable Dynamic Jitter Buffer
- Bad Frame Interpolation
- Gain/Attenuation Settings

Provisioning and Configuration

- SIP (RFC3261) compliance
- LCD configuration password protection
- Provide Proxy Mode or Peer-to-Peer Mode (Non Proxy Server needed) selection
- Ring tone, Speaker and Handset volume adjustable
- Dial path selection (PSTN or IP mode)
- Support DNS server inquiry

Management Features:

- Software Upgrade: TFTP/FTP download
- Three easy ways for system configuration
 - LCD Front Panel
 - Web Browser
 - TELNET

Certification

- CE, 3C

3. Physical specification & Environment

Dimension:

- 215mm(W) x 71mm(H) x 198mm(D)

Weight:

- 834 grams (unit)

Power Supply:

- DC 9V output adaptor, AC 120Vac input

Operation Environment:

- Humidity: 10 to 90 % (Non-condensing)
- Operational Temperature: 0 to +40 °C

Storage Environment:

- Humidity: 10 to 90 % (Non-condensing)
- Storage Temperature: -10 to +50 °C

Analog Specification:

- 2 wires (RJ-11) *1 (Voltage: -48V)
- AC Impedance 600

Chapter 2 Configuring the SIP-Phone through LCD Phone menus

Note:

1. After any configuration is made for the SIP-Phone, user has to do **Reboot** in the selection 7 “Reboot”.
2. We suggest user to set IP address via LCD menu 5 2.3.4.5 first, then go to chapter 3 to do other detail configurations via web browser.

1. Initialize SIP-Phone

1. When power on the SIP-Phone, the LCD screen shows as below. Now SIP-Phone is running Boot sector program.

IP-Phone Board Start Booting

2. When SIP-Phone finishes boot program initialization. User can see flashing greeting as below:

System Initializing.....

3. Then SIP-Phone get into standby mode:

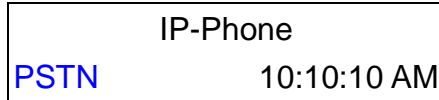
SIP-Phone
Proxy 10:10:10 AM

The main LCD screen would be shown as similar as above. “Proxy” means the SIP-Phone is in Proxy Mode, and when SIP-Phone is connected to SNTP server, on LCD will show current time captured from SNTP server.

4. When SIP-Phone is under peer-to-peer mode, on LCD will show “P2P” instead of “Proxy”.

SIP-Phone
P2P 10:10:10 AM

5. After pressing the **PSTN** button, the “Proxy” or “P2P” will be replaced by “PSTN”. Please notice that user must plug PSTN line in RJ-11 port when SIP-Phone is in PSTN mode. SIP Phone will always stay in IP mode, after a PSTN call is finished, SIP Phone will automatically return to IP mode..



6. Press **←** or **→** to enter configuration mode then press **OK** button to enter sub menus; press **C** can jump out current menu to previous level.
1. Call List
 2. Forward Type
 3. Phone Book
 4. Ringer Settings
 5. Network
 6. Advanced Settings (protected by password)
 7. Reboot

2. LCD Menu Configuration

User can set the following configurations by LCD keypad.

Note:

1. Press **TRANSFER** before input data can switch characters to be capital or lowercase.
2. Press **SPEED** before input data can switch input mode as character mode or digit mode; for example, user wants to enter IP address, after pressing **SPEED** can enter digits directly.
3. When user is inputting data, press **C** will jump out to menu list, press **←** will clear previous input data.

1. Call List

If there is an unanswered IP call, it will be kept in message box. **MESSAGE** LED will be flashing until user press **MESSAGE** to check miss call and re-press MESSAGE to return to main screen.

- (1) **Missed Calls** : to see all missed calls in message box.
- (2) **Received Calls** : to see all received calls in message box.
- (3) **Dialed Numbers**: to see all dialed calls in message box.

2. Forward Type

There are 3 selections in Forward type, user must select under which condition to forward calls.

(1) Busy

When SIP-Phone is in busy status, the incoming call will be forwarded to the assigned phone number.

A. Activate

Enter a forwarded phone number to activate busy forward function.

B. Deactivate

Deactivate Busy Forward function.

(2) No Response

When SIP-Phone hasn't been picked up for around 10 seconds, the incoming call will be forwarded to the assigned phone number.

A. Activate

Enter a forwarded phone number to activate no response forward function.

B. Deactivate

Deactivate No Response Forward function.

(3) Unconditional

It is included the above two types. Whether the SIP-Phone is in which status, calls will be automatically forwarded to the assigned phone number.

A. Activate

Enter a forwarded phone number to activate Unconditional Forward function.

B. Deactivate

Deactivate Unconditional Forward function.

3. Phone Book

1. List

List all records of name, telephone number, and IP address in the phone address book.

2. Edit/Delete

Edit or delete a record of name, telephone number, and IP address of the phone address book.

3. New

Add a new record of name, telephone number, and IP address of the phone address book.

4. Ringer Settings

1. Volume

User can adjust ring volume by press **←** or **→** on the keypad to decrease or increase ringer volume.

2. Style

There are three tone styles for SIP-Phone. Move the ">" symbol by press **←** or **→** on the keypad to select the tone style preferred, then press **OK** to confirm it.

5. Network

1. Information

User can press ← or → to check current network status:

- (1) Mode: Display current network connection mode of SIP-Phone to be Static (Fixed IP), DHCP, or PPPoE.

Note:

When SIP-Phone is under DHCP mode, then change to Static mode, the following items: IP address, Subnet Mask, Default Gateway, will display empty, after reboot, user can see information again.

- (2) IP, Mask, Gateway: display current IP information.

2. Network Mode

Set network mode of SIP-Phone to be Static (Fixed IP), DHCP, or PPPoE.

3. IP address

Set IP address of SIP-Phone.

4. Subnet Mask:

Set subnet mask address of SIP-Phone.

5. Default Gateway

Set default gateway address of SIP-Phone.

6. Domain Name Server

Set IP address of Domain Name Server. Once SIP-Phone can connect to DNS server, user can set URL address for Proxy server or Phone book instead of IP address.

7. PPPoE Configuration

(1) User Name

Set PPPoE connection authentication user name.

(2) Password

Set PPPoE connection authentication password.

(3) Auto Re-connect

Choose ON or OFF to enable or disable this function. If user enables this function, after PPPoE disconnected, SIP-Phone will automatically reboot to re-connect, and after reboot, if SIP-Phone still can't connect with server, SIP-Phone will keep trying to connect. On the other hand, if user disables this function, SIP-Phone won't reboot and keep trying to connect.

8. SNTP Configuration

(1) SNTP Mode:

User can set SNTP function to be on or off, which means SIP-Phone will capture current time from SNTP server or not.

(2) SNTP Server:

User can specify a SNTP server for SIP-Phone to capture current time.

(3) Time Zone:

User can set time zone via pressing ← or → according to the location SIP-Phone is. For example, in Taiwan the time zone should be set as GMT+8:00.

9. Behind IP-Sharing

(1) If SIP-Phone is behind IP sharing or NAT device, on IP sharing must enable “**DMZ**” function or set “**Virtual Server**” to open ports (UDP port: 5060 and 16384).

(2) User must enter public IP address of IP sharing.

6. Advanced Settings (protected by password)

● Please Enter Password:

User must key in password to enter this menu, selections under this command are all important ones, which can only be configured by advanced users.

Note:

1. Default Password is **empty**, user can enter this sub-menu directly, once password has been set in **3.LCD Menu password**, user must input password before enter this sub-menu.
2. If user forget password, please contact with the distributor, we will generate a specific password according to your MAC address of SIP-Phone.
3. User can also try to configure SIP-Phone via Telnet or Web browser with default IP address: 10.1.1.3. (This only works when default IP address hasn't been changed.)

1. SIP Settings

(1) Connect Mode

Select SIP connection mode to be peer-to-peer mode or Proxy mode.

(2) Proxy

A. Proxy

Set Proxy IP address or Domain Name.

B. Outbound proxy

Set Outbound Proxy IP address or Domain Name.

C. Proxy port

Set Proxy port for SIP-Phone to send messages.

D. Expire (in seconds)

Set expire time of registration, in the duration of 2/3 expire time, SIP-Phone will re-register to Proxy Server again.

(3) User Info

A. User Name

Set User Name of SIP-Phone to register to Proxy Server. If Proxy server doesn't request specific User name, please enter Line number here.

B. Line Number

Set Line Number of SIP-Phone to register to Proxy Server.

C. Password

Set User Password of SIP-Phone to register to Proxy Server. This configuration is not necessary, if Proxy server doesn't request client to set password, user only has to set User Name the same as Line Number.

2. Firmware Update

(1) Download method

There are two methods to download new version file, please move the ">" symbol by press **←** or **→** on the keypad to select TFTP or FTP method, then press **OK** to confirm it.

(2) FTP/TFTP Sever

User has to offer one TFTP/FTP server IP Address and set this IP Address via keypad. The IP Address is necessary for upgrading SIP-Phone new application rom file.

(3) FTP Account

User has to input user name for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

(4) FTP Password

User has to input password for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

(5) File Name

User has to press the file name of new application rom file prepared for upgrading

(6) Application Version

Show current version of application.

(7) Start to Upgrade

Select YES or NO to start upgrade.

Note:

Download via LCD command can only upgrade new ***application*** rom file.

4. LCD Menu Password

Set entry password of phone LCD menu.

7. Reboot

Reboot machine. It is necessary and important for user to reboot SIP-Phone after any configurations has been made. SIP-Phone will ask user again before reboot.

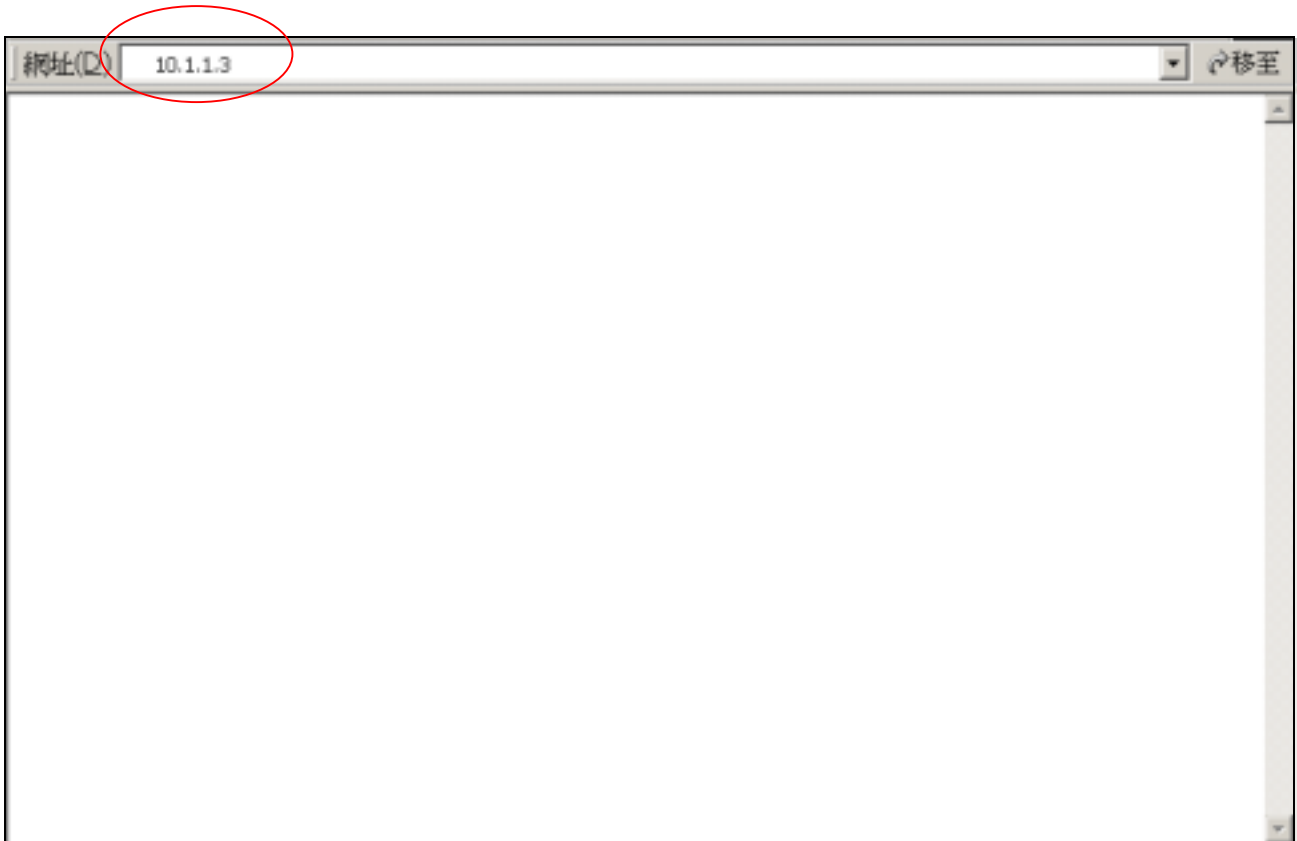
Chapter 3 Configuring the SIP-Phone through Web Pages

The HTTPD web management interface provides user an easier way to configure rather than command line method through TELNET.

The configuration function and steps are similar with the way through command line. Please refer to the chapter 4-Configuring the SIP-Phone through Telnet command lines for more detail information. Below is a guide for user to configure via web interface.

Step 1. Browse the IP Address predefined via Keypad

Please enter IP address (user have to set via LCD menu first) of SIP-Phone in web browser. If user failed to set IP address via LCD menu, the **default IP address of SIP-Phone is 10.1.1.3**, user can try to connect to SIP-Phone via this default IP via web interface.



Step 2. Input the login name and password

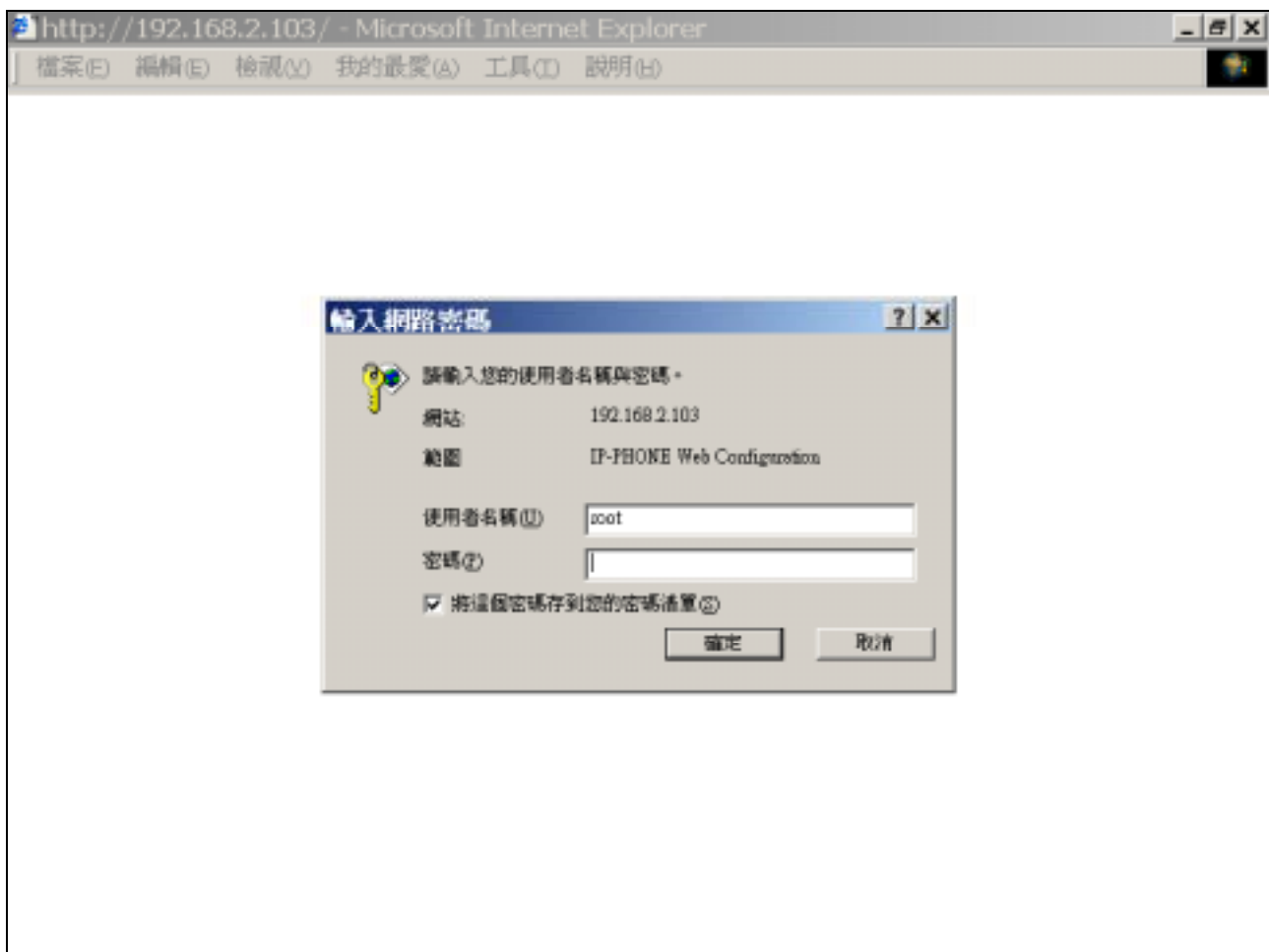
- Login name: root / user

Note:

Login with “user” only has authority as below:

1. Modify network configuration
2. Modify Phone Book
3. Change login password of “user”
4. Reboot

- Password (The same with TELNET): Null (just press confirm, no need to key in password in default value)



Note: User can set password later in **8.PASSWORD** via web interface.

Step 3. Enter the web interface main screen

After enter login name and password, user can see web interface main screen as below.



Step 4. Start to configure

Most important items are [Network Interface](#), [SIP Information](#), and [Phone Book \(in Peer-to-Peer mode\)](#). Please remember to configure these commands before start to work with SIP-Phone.

Note:

After change any settings, please remember to **reboot** (in [Reboot System](#)) SIP-Phone so that changes can take effect.

1. Network Interface

Please refer to chapter 4.7 [ifaddr] command.

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'Network Interface' configuration window on the right. The 'Network Interface' window contains the following fields and options:

Network Interface	
IP Address:	10 . 1 . 1 . 3
Subnet Mask:	255 . 0 . 0 . 0
Default routing gateway:	10 . 1 . 1 . 254
Get IP Mode:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP <input type="radio"/> PPPoE
SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	8
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
UPnP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	0 . 0 . 0 . 0
Primary DNS Server:	168 . 95 . 192 . 1
Secondary DNS Server:	168 . 95 . 1 . 1
OK	

- **IP Address:** Set IP Address of SIP-Phone
- **Subnet Mask:** Set the Subnet Mask of SIP-Phone
- **Default routing gateway:** Set Default routing gateway of SIP-Phone
- **Get IP Mode:** User has to set SIP-Phone to use which network mode.
 1. **Fixed IP:** User has to assign a fixed IP to SIP-Phone.
 2. **DHCP:** When DHCP function enables, SIP-Phone will automatically search DHCP server after reboot.
 3. **PPPoE:** If SIP-Phone is working with PPPoE connection, user have to set related parameters in “**PPPoE Configure**” page.

Note:

If User set “Get IP mode” as DHCP or PPPoE, IP address, Subnet Mask, and Default routing gateway will become 0.0.0.0 and not allow to be configured.

SIP-Phone Configuration Menu

- [Network Interface](#)
- [SIP Information](#)
- [System Configuration](#)
- [PPPoE Configure](#)
- [Voice Setting](#)
- [Phone Book](#)
- [DSCP Configuration](#)
- [Password](#)
- [ROM Configuration](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

Network Interface

IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Subnet Mask:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Default routing gateway:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Get IP Mode:	<input type="radio"/> Fixed IP <input checked="" type="radio"/> DHCP <input type="radio"/> PPPoE
SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
SNTP Server Address:	<input type="text"/> 168 . <input type="text"/> 95 . <input type="text"/> 195 . <input type="text"/> 12
GMT:	<input type="text"/> 8
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
UPnP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	<input type="text"/> 0 . <input type="text"/> 0 . <input type="text"/> 0 . <input type="text"/> 0
Primary DNS Server:	<input type="text"/> 168 . <input type="text"/> 95 . <input type="text"/> 1 . <input type="text"/> 1
Secondary DNS Server:	<input type="text"/> 168 . <input type="text"/> 95 . <input type="text"/> 1 . <input type="text"/> 2

OK

SIP-Phone Configuration Menu

- [Network Interface](#)
- [SIP Information](#)
- [System Configuration](#)
- [PPPoE Configure](#)
- [Voice Setting](#)
- [Phone Book](#)
- [DSCP Configuration](#)
- [Password](#)
- [ROM Configuration](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

Network Interface

IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Subnet Mask:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Default routing gateway:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Get IP Mode:	<input type="radio"/> Fixed IP <input type="radio"/> DHCP <input checked="" type="radio"/> PPPoE
SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
SNTP Server Address:	<input type="text"/> 168 . <input type="text"/> 95 . <input type="text"/> 195 . <input type="text"/> 12
GMT:	<input type="text"/> 8
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
UPnP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	<input type="text"/> 0 . <input type="text"/> 0 . <input type="text"/> 0 . <input type="text"/> 0
Primary DNS Server:	<input type="text"/> 168 . <input type="text"/> 95 . <input type="text"/> 1 . <input type="text"/> 1
Secondary DNS Server:	<input type="text"/> 168 . <input type="text"/> 95 . <input type="text"/> 1 . <input type="text"/> 2

OK

- **SNTP:** Enable / Disable the Simple Network Time Protocol function
- **SNTP Server Address:** Set SNTP Server Address
When SNTP server is available, enable SIP-Phone SNTP function to point to SNTP server IP address so that SIP-Phone can get correct current time.
- **GMT:** Set time zone for SNTP Server time
User can set different time zone according to the location of SIP-Phone. For example, in Taiwan the time zone should be set as 8, which means GMT+8.
- **IP Sharing:** Enable it if SIP-Phone is behind IP Sharing router.
- **UPnP:** Enable it if IP sharing or NAT device supports UPnP function so that no need to configure IP sharing or SIP-Phone when SIP-Phone is behind NAT device.
- **IP Sharing Server Address:** Set Public IP Address of IP Sharing router for SIP-Phone to work behind IP sharing.
- **Primary DNS Server:** Set Primary Domain Name Server IP address.
User can set Domain Name Server IP address. Once SIP-Phone can connect with DNS server, user can specify URL address instead of IP address for Proxy and phone book IP address.
- **Secondary DNS Server:** Set Secondary Domain Name Server IP address.

2. SIP Information

Please refer to chapter 4.13 [SIP] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'SIP Configuration' form on the right. The menu includes links for Network Interface, SIP Information (selected), System Configuration, PPPoE Configure, Voice Setting, Phone Book, DDNS Configuration, DSCP Configuration, Password, ROM Configuration, Flash Clean, and Reboot System. The SIP Configuration form contains the following fields:

SIP Configuration	
Run Mode:	<input type="radio"/> Peer-2-Peer <input checked="" type="radio"/> Proxy
Proxy IP Address:	<input type="text" value="10.1.1.2"/>
Outbound Proxy:	<input type="text" value="null"/>
Proxy port:	<input type="text" value="5060"/>
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	<input type="text" value="null"/>
Line Number:	<input type="text" value="1001"/>
Line Account:	<input type="text" value="1001"/>
Line Password:	<input type="text" value="***"/>
SIP port:	<input type="text" value="5060"/>
RTP Port:	<input type="text" value="16384"/>
Expire:	<input type="text" value="60"/>
<input type="button" value="OK"/>	

- **Run Mode:** Select SIP-Phone to work under Peer-to-Peer mode or Proxy mode.
- **Proxy IP Address:** Set Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Interface**).
- **Outbound Proxy:** Set IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Configure**) of outbound Proxy server.
- **Proxy port:** Set Proxy port for SIP-Phone to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- **Phone Book Search:** enable/disable phone book search function. If user enabled this function, SIP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, SIP-Phone will make call to related IP address.
- **Prefix String:** set prefix string. If user ID contains alphabets, user can set it as prefix

string here. For example, if Account Name is 123, SIP-Phone will send out messages as Account Name @"IP address of Proxy", if user sets prefix as abc, SIP-Phone will send out as abc123@"IP address of Proxy". This function is for special proxy server.

- **Line Number:** identify one number for the SIP-Phone to register to the Proxy.
- **Line Account:** set user name of SIP-Phone for registering. User can set user name and password for registering. If password is not needed, please set user name the same as line number or SIP-Phone won't register successfully.
- **Line Password:** set password for registering.
- **SIP Port:** set SIP UDP port.
- **RTP Port:** set RTP port for sending voice data.
- **Expire:** set expire time of registration. SIP-Phone will keep re-registering to proxy server before expire times out

3. System Configuration

Please refer to chapter 4.12 [sysconf] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left sidebar and the 'System Configuration' tab on the right. The sidebar menu includes links for Network Interface, SIP Information, System Configuration (highlighted), PPPoE Configure, Voice Setting, Phone Book, DDNS Configuration, DSCP Configuration, Password, ROM Configuration, Flash Clean, and Reboot System. The 'System Configuration' tab contains the following settings:

System Configuration	
Keypad DTMF Type:	<input checked="" type="radio"/> In-Band <input type="radio"/> RFC2833 <input type="radio"/> INFO
RFC2833 Payload Type:	<input type="text" value="96"/>
Inter Digit Time:	<input type="text" value="3"/>
End of Dial Digit:	<input type="radio"/> NONE <input type="radio"/> * <input checked="" type="radio"/> #
<input type="button" value="OK"/>	

- **Keypad DTMF Type:** set DTMF type. User can select DTMF type SIP-Phone transmits.
- **RFC2833 Payload Type:** change RFC2833 Payload type. This is for special request from the other site, if RFC2833 payload types of 2 sites are different, it may cause some problem of connection.
- **Inter Digit Time:** Set the DTMF inter digit time (second)
To set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, SIP-Phone will dial out all number pressed.
- **End of Dial Digit:** select end of dialing key, e.g. set end of dial key as * button, after finished pressing dialing number then press * will dial out.

4. PPPoE Configure

Please refer to chapter 4.11[pppoe] command

The screenshot shows a web-based configuration interface for a SIP-Phone. On the left is a 'SIP-Phone Configuration Menu' with links: Network Interface, SIP Information, System Configuration, PPPoE Configure (highlighted), Voice Setting, Phone Book, DDNS Configuration, DSCP Configuration, Password, ROM Configuration, Flash Clean, and Reboot System. The main area is titled 'PPPoE Device Information and Configuration'. It contains the following fields and controls:

PPPoE Device Information and Configuration	
User Name:	<input type="text" value="pppoe"/>
Password:	<input type="password" value="*****"/>
Reboot After Remote Host Disconnection:	<input checked="" type="radio"/> On <input type="radio"/> Off
PPPoE Echo:	<input checked="" type="radio"/> enable <input type="radio"/> disable
Authenticate:	<input type="text"/>
Protocol:	<input type="text" value="TCP/IP"/>
Device:	<input type="text" value="PPP/PPPoE"/>
<input type="button" value="OK"/>	

- **User Name:** Set PPPoE authentication User Name.
- **Password:** Set PPPoE authentication password.
- **Reboot After Remote Host Disconnection:** Enable/Disable auto reboot after PPPoE disconnection
If user enables this function, after PPPoE being disconnected, SIP-Phone will automatically reboot to re-connect, and after reboot, if SIP-Phone still can't get contact with server, SIP-Phone will keep trying to connect. After re-connected, SIP-Phone will also restart system. On the other hand, if user disables this function, SIP-Phone won't reboot and keep trying to connect.
- **PPPoE Echo:** Enable or Disable PPPoE Echo function. If user enabled this feature, SIP Phone will send out echo request to check PPPoE connection status. Please notice that if user disables this function, SIP Phone cannot detect if PPPoE connection is still alive or not.
- **Other items:** for reference only, cannot allow to be configured.

5. Voice Setting

Please refer to chapter 4.15 [voice] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'Voice Setting' configuration page on the right. The 'Voice Setting' page includes fields for Codec Priority (1st to 5th), Frame Size, G.723 Silence Suppression, Volume (voice, ring, input, DTMF), Echo Cancelor, Jitter Buffer (Min. Delay, Max. Delay), and Optimized Factor (Jitter). An 'OK' button is located at the bottom right of the configuration area.

Voice Setting					
Codec Priority	1st G.723.1	2nd G.729	3rd G.711mu-Law	4th G.711A-Law	5th G.729
Frame Size	G.723.1 60ms	G.729a 60ms	G.729 60ms	G.711mu 40ms	G.711A 40ms
G.723 Silence Suppression:	<input type="radio"/> enable <input checked="" type="radio"/> disable				
Volume:	voice 32	ring 32	input 32	DTMF 31	
Echo Cancelor:	<input checked="" type="radio"/> enable <input type="radio"/> disable				
Jitter Buffer:	Min. Delay 90		Max. Delay 150		
Optimized Factor (Jitter):	9				
OK					

- **Codec Priority:** set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need. For example, user can only set first priority as G.723.1, and set the others as x, that means only G.723.1 is available.
- **Frame Size:** User can set different packet size for each codec.
- **G.723 Silence Suppression:** Enable / Disable sound compression and comfort noise generation. It is only for codec G.723.1
- **Volume:** Adjust the volume in "Voice" (sending out); "Input" (receiving); "DTMF" (DTMF sending out).
- **Echo Cancelor:** Enable / Disable (suggested always Enable this function).
- **Jitter Buffer:** Set Min. Delay and Max. Delay of Jitter Buffer for voice packets.
- **Optimized Factor (Jitter):** Set Optimized Factor of voice, this is for advanced user only, please contact with your distributor before making any change.

6. Phone Book

Please refer to chapter 4.10 [pbook] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left sidebar, with the 'Phone Book' option selected. The main content area is divided into two sections: 'Phone Book' and 'New Record'.

Phone Book

Index	Name	IP_Address	e164

New Record

Index	Name	IP Address	E164 No.
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

Below the 'New Record' table are two buttons: 'Add Data' and 'Delete Data'.

- **Add Data:** User can specify 20 sets of phone book via web interface. Please input index, Name, IP Address and E.164 number of the destination device.
- **Delete Date:** User can delete any configured phone book data by index.

7. DDNS Configuration

Please refer to chapter 4.17 [ddns] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'DDNS Configuration' form on the right. The menu includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DDNS Configuration (highlighted), DSCP Configuration, Password, ROM Configuration, Flash Clean, and Reboot System. The DDNS Configuration form contains the following fields and controls:

DDNS Configuration	
Status:	<input type="radio"/> Enable <input checked="" type="radio"/> Disale
Server:	<input type="text" value="www.dyndns.org"/>
Host Name:	<input type="text"/>
ID:	<input type="text"/>
Password:	<input type="text"/>
Check IP:	<input type="radio"/> On <input checked="" type="radio"/> Off
IP Check Server 1:	<input type="text" value="checkip.dyndns.org"/>
IP Check Server 2:	<input type="text" value="checkip.dyndns.org"/>
Check every:	<input type="text" value="0"/> <input type="radio"/> mintues <input type="radio"/> hours <input checked="" type="radio"/> Off
<input type="button" value="DONE"/>	

- **Status:** to enable/disable DDNS function
- **Server:** to choose one DDNS server, on which user has already registered. (Now only one DDNS server is available---www.dyndns.org)
- **Host Name:** to set the registered Domain Name of SIP Phone
- **ID:** to set login ID of registered account to log in DDNS server
- **Password:** set password of registered account to log in DDNS server
- **Check IP:** to enable/disable check IP function. If SIP Phone is behind IP sharing, when this function is enabled, SIP Phone will check it's public IP address by asking IP address check server and send to DDNS server to update DDNS data. If this function is disabled, when SIP Phone is behind IP sharing, it will send it's private IP address to DDNS server
- **IP Check Server 1:** to set IP address check server
- **IP Check Server 2:** to set secondary IP address check server
- **Check every /minutes /hours /off:** to set the update interval time. SIP Phone will re-update its IP address in this time.

8. DSCP Configuration

Please refer to chapter 4.16 [tos] command

SIP-Phone Configuration Menu

- [Network Interface](#)
- [SIP Information](#)
- [System Configuration](#)
- [PPPoE Configure](#)
- [Voice Setting](#)
- [Phone Book](#)
- [DSCP Configuration](#)
- [Password](#)
- [ROM Configuration](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

DiffServ Code Point(DSCP) Configuration

=== Signal Packet ===

☐ Assured Forwarding (AF) PHB Delay Priority : Drop Precedence :

☐ Expedited Forwarding(EF) PHB

☒ Default

☐ User Assign Special DSCP Code:

=== RTP Packet ===

☐ Assured Forwarding (AF) PHB Delay Priority : Drop Precedence :

☐ Expedited Forwarding(EF) PHB

☒ Default

☐ User Assign Special DSCP Code:

Set Signal or RTP Packet DSCP value:

- **Assured Forwarding (AF) PHB:** Select Delay priority and Drop Precedence
- **Expedited Forwarding (EF) PHB:** Select TOS value as EF
- **Default:** Select TOS value as 0
- **User Assign Special DSCP Code:** User can set other unspecified value here.

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority. As per the definition in RFC2474, the DS field is Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined in RFC2597 to classify the traffic into different service classes. The mapping of Code Point value of DS-field to egress traffic priorities is shown as follows.

DROP Precedence	Class #1	Class #2	Class #3	Class #4
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

Please refer to RFC standard documents for more information about what is DSCP.

9. Password

Please refer to chapter 4.19 **[password]** command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'Password' configuration window on the right. The menu includes options like Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password (highlighted), ROM Configuration, Flash Clean, Commit Data, and Reboot System. The Password window has a title bar 'Password' and contains a dropdown menu for login name (currently 'root'), three input fields for 'Current Password:', 'New Password:', and 'Confirm New Password:', and two buttons at the bottom: 'CHANGE' and 'ABORT'.

Password	
<div>root</div>	Current Password: <input type="text"/>
	New Password: <input type="text"/>
	Confirm New Password: <input type="text"/>
<div>CHANGE</div> <div>ABORT</div>	

- **Change:** First select login name as root or user, then enter current password, new password and confirm new password again to set new password.
- **Abort:** Press abort will clean all inputs.

10. Rom Configuration

Please refer to chapter 4.18 [rom] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'ROM Configuration' form on the right. The menu includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration (highlighted), Flash Clean, Commit Data, and Reboot System. The ROM Configuration form contains the following fields:

ROM Configuration	
FTP/TFTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Target File name:	<input type="text"/>
Method:	<input type="text" value="TFTP"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	<input type="text" value="Application Image"/>
<input type="button" value="OK"/>	

- **FTP/TFTP Server IP Address:** Set TFTP server IP address
- **Target File name:** Set file name prepared to upgrade
- **Method:** Select download method as TFTP or FTP
- **FTP Login:** Set FTP login name and password
- **Target File Type:** Select which sector of SIP-Phone to upgrade

Note:

1. After 2mb file download is finished, all configurations might change to default values, user has to configure again.
2. After upgrade Application, please remember to execute Flash Clean, which will clean all configurations become factory values except IP address.

11. Flash Clean

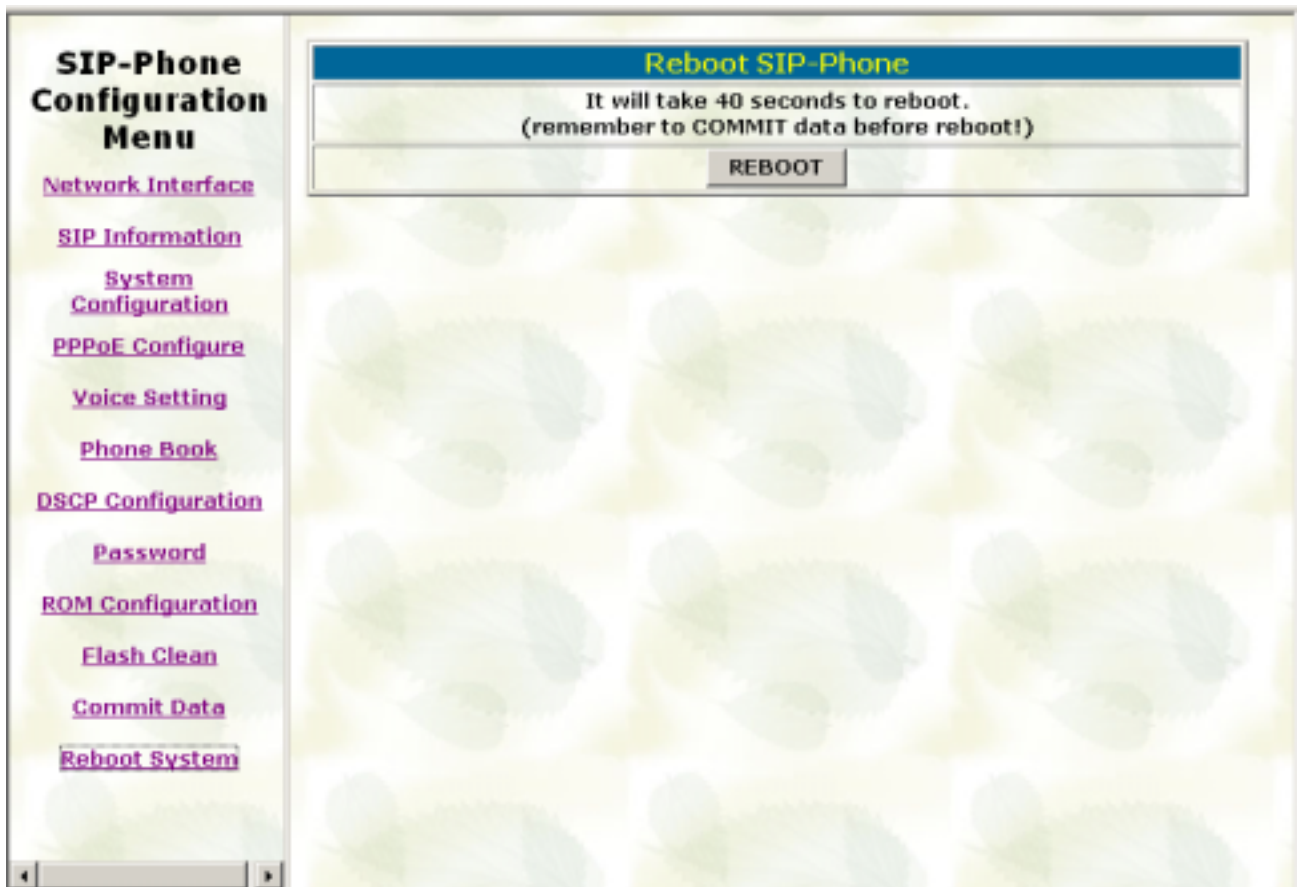
Please refer to 4.15 [flash] command.



- Press CLEAN will clean all configurations of SIP-Phone and reset to factory default value.

Note: User must re-configure all commands all over again (except Network Configure) once execute this function,

12. Reboot System



- Press reboot will reset SIP-Phone.

Note: To execute reboot via web browser, SIP-Phone will automatically save all data before reboot. To execute reboot via TELNET command, please remember to do **Commit Data** before **Reboot System**.

Chapter 4 Configuring the SIP-Phone through Telnet command lines

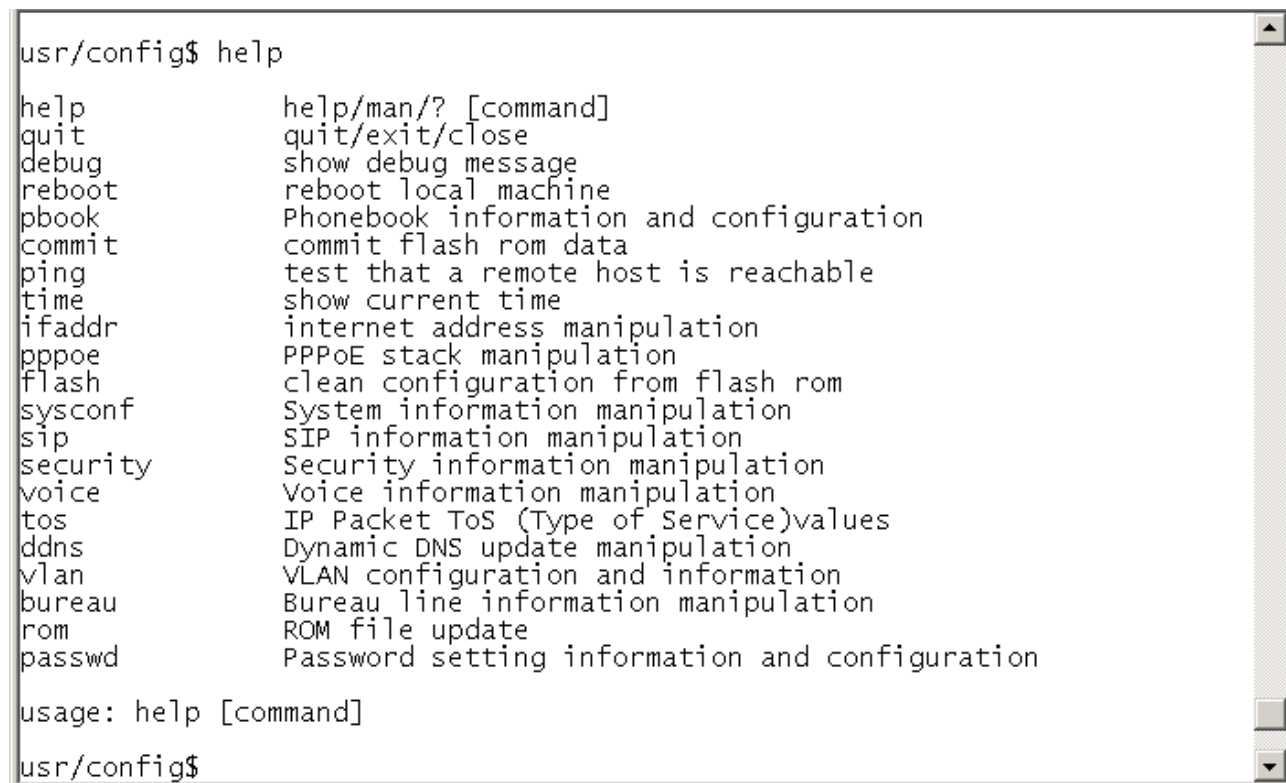
After setting the IP Address of SIP-Phone and reboot, (please refer to LCD Menu: 5-3.4.5), user can enter into Telnet command lines.

Note:

1. After user enter SIP-Phone configuration via telnet, please use login: "**root**", password: null, press enter to enter command lines. If user forgets password, please contact with the distributor, we will generate a specific password according to your MAC address of SIP-Phone.
2. User must input lower-case command, but contents of configurations such as SIP alias or user name etc, user can set as capital case.
3. After any change of configuration, please remember to do **commit** command to save changes and then **reboot** command to reboot system.

1. [help] command

Type **help** or **man** or **?** to display all the command lists. The following figure is shown all commands of SIP-Phone.

A screenshot of a telnet session window. The prompt is 'usr/config\$'. The user has entered 'help'. The output is a list of commands and their descriptions. The list includes: help, quit, debug, reboot, pbook, commit, ping, time, ifaddr, pppoe, flash, sysconf, sip, security, voice, tos, ddns, vlan, bureau, rom, passwd, and usage. Each command is followed by a brief description of its function. The window has a scrollbar on the right side.

```
usr/config$ help
help          help/man/? [command]
quit          quit/exit/close
debug         show debug message
reboot        reboot local machine
pbook         Phonebook information and configuration
commit        commit flash rom data
ping          test that a remote host is reachable
time          show current time
ifaddr        internet address manipulation
pppoe         PPPoE stack manipulation
flash         clean configuration from flash rom
sysconf       System information manipulation
sip           SIP information manipulation
security      Security information manipulation
voice         Voice information manipulation
tos           IP Packet ToS (Type of Service)values
ddns          Dynamic DNS update manipulation
vlan          VLAN configuration and information
bureau        Bureau line information manipulation
rom           ROM file update
passwd        Password setting information and configuration

usage: help [command]
usr/config$
```

2. [quit] command

Type **quit/exit/close** will logout SIP-Phone and Telnet Program.

3. [debug] command

This command is for engineers to debug system of SIP-Phone. User can add debug flag via command **debug –add “debug flags”**, and then start debug function via command **debug –open**. When SIP-Phone is working on screen will display related debug messages. Most frequently used debug flag are “sip”, “fsm”, “msg”...etc.



```
usr/config$ debug
debug message information and configuration
Usage:
debug [-add type1 [[type2]...]] | -open | -close | -status
    -status      Display the enabled debug flags.
    -add         Add debug flag.
    -delete      Remove specified debug flag.
    -open        Start to show debug messages.
    -close       Stop showing debug messages.
Example:
debug -add sip msg
debug -open
usr/config$
```

4. [reboot] command

After typing **commit** command, type **reboot** to restart the SIP-Phone.

Sometimes after user type reboot, on terminal screen will display: "Data modified, commit to flash rom?" which means SIP-Phone will record call history or not. (Ex. REDIAL, outgoing and incoming call data)

5. [pbook] command

This command is functional both in Proxy mode and Peer-to-Peer mode. In proxy mode, use speed dial or 10 DL button will dial out e.164 number in phone book. In the other hand, in peer-to-peer mode, SIP-Phone will dial out IP address.

1. **-print**: display phone book data. User can print all data in phone book by command (**pbook –print**). Furthermore, user can also print only a section of data by indicate parameter “start index” and “end index” (**pbook –print “start index” “end index”**). If parameter “end index” is omitted, only record “start index” will be displayed. (**pbook –print “start prefix”**).
2. **-add**: add a new record in phone book table by giving name, IP address, and e.164 number of callee endpoint.
(**pbook –add name “X” ip “xxx.xxx.xxx.xxx” e164 “X”**)
3. **-search**: search any record in the phone book according to IP address, name or

e164 number

4. **-delete**: delete a record of certain listed index in phone book table. (**pbook -delete "index number"**)
5. **-insert**: insert a record in specified index of phone book.
6. **-modify**: modify record of a certain index in phone book. Please notice that the name, IP address and e164 number must be modified together; user cannot just modify one parameter only.

(pbook -modify "index" name "X" ip "xxx.xxx.xxx.xxx" e164 "X")

```
usr/config$ pbook
phonebook information and configuration
Usage:
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-search [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-insert [index] [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-delete [index]]
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenumber]]

    -print      Display phonebook data.
    -add        Add an record to phonebook.
    -search     Search an record in phonebook.
    -delete     Delete an record from phonebook.
    -insert     Insert an record to phonebook in specified position.
    -modify     Modify an exist record.

Note:
If parameter 'end_record' is omitted, only record 'start_record' will be display.
If both parameters 'end_record' and 'start_record' are omitted, all records will be display.
Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
Range of index setting value (1~100),
Example:
pbook -print 1 10
pbook -print 1
pbook -print
pbook -add name Test ip 210.59.163.202 e164 1001
pbook -insert 3 name Test ip 210.59.163.202 e164 1001
pbook -delete 3
pbook -search ip 192.168.4.99
pbook -modify 3 name Test ip 210.59.163.202 e164 1001
usr/config$
```

6. [commit] command

Save any changes after configuring the SIP-Phone.

```
usr/config$ commit

this may take a few seconds, please wait....
Commit to flash memory ok!
usr/config$
```

7. [ping] command

Command **ping** can test which the IP address is reachable or not.

Usage: **ping "IP address"**

The message will display packets transmitting condition or no answer from the IP address.

```

usr/config$ ping
usr/config$ ping 192.168.2.107
PING 192.168.2.107: 56 data bytes
64 bytes from 192.168.2.107: icmp_seq=0. time=5. ms
64 bytes from 192.168.2.107: icmp_seq=1. time=0. ms
64 bytes from 192.168.2.107: icmp_seq=2. time=0. ms
64 bytes from 192.168.2.107: icmp_seq=3. time=0. ms
----192.168.2.107 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 0/1/5
usr/config$ ping 192.168.99.99
PING 192.168.99.99: 56 data bytes
no answer from 192.168.99.99
usr/config$

```

8. [time] command

When SIP-Phone enables SNTP function and be able to connect with SNTP server, type **time** command will show the current time retrieved from SNTP server.

```

usr/config$ time
Current time is TUE FEB 03 14:50:08 2004
usr/config$

```

9. [ifaddr] command

Configure and display the SIP-Phone IP information.

1. **-print:** print out all current configurations of ifaddr command.
2. **-ip, -mask, -gate:** Set SIP-Phone IP Address, subnet mask and default gateway respectively.
5. **-ipmode:** Set SIP-Phone network mode to be Fixed IP, DHCP or PPPoE.
 When User set IP mode to be fixed IP, please set IP, subnet Mask, default gateway as mentioned in item 2.
 If User set IP mode to be DHCP, SIP-Phone will search for DHCP server to capture IP address after reboot.
 If user set IP mode to be PPPoE, please remember to set related parameters under **[pppoe] command**.
6. **-sntp:** When SNTP server is available, enable SIP-Phone SNTP function and assign SNTP server IP address so that SIP-Phone can capture current time from SNTP server. (**ifaddr -sntp 1 "xxx.xxx.xxx.xxx"**)
7. **-dns:** User can set Domain Name Server IP address. Once SIP-Phone can connect with DNS server, user can specify URL address instead of IP address for Proxy Server and phone book IP address...etc.
8. **-timezone:** User can set different time zone according to the location SIP-Phone is. For example, in Taiwan the time zone should be set as 8, which means GMT+8. (GMT-8: **ifaddr -timezone -8**)

9. **-ipsharing:** If SIP-Phone is behind an IP-sharing, user must enable IP sharing function and specify public IP address. (**ifaddr -ipsharing 0/1 “public IP address of IP sharing”**, 0 for disable and 1 for enable)

Note:

Some Proxy servers support endpoint behind NAT function, in this case SIP-Phone doesn't have to enable IP sharing function, please contact with your Proxy Server vendor.

10. **-upnp:** enable/disable UPnP function. If the IP sharing or NAT device supports UPnP, user can enable UPnP function so that SIP-Phone will automatically connect with NAT device without configuration in SIP Phone and NAT device.
11. **-server:** set EMS server IP address.
12. **-id:** set EMS login user ID.
13. **-pwd:** set EMS login user password.
14. **-emstime:** set EMS server recycle time.

```
usr/config$ ifaddr

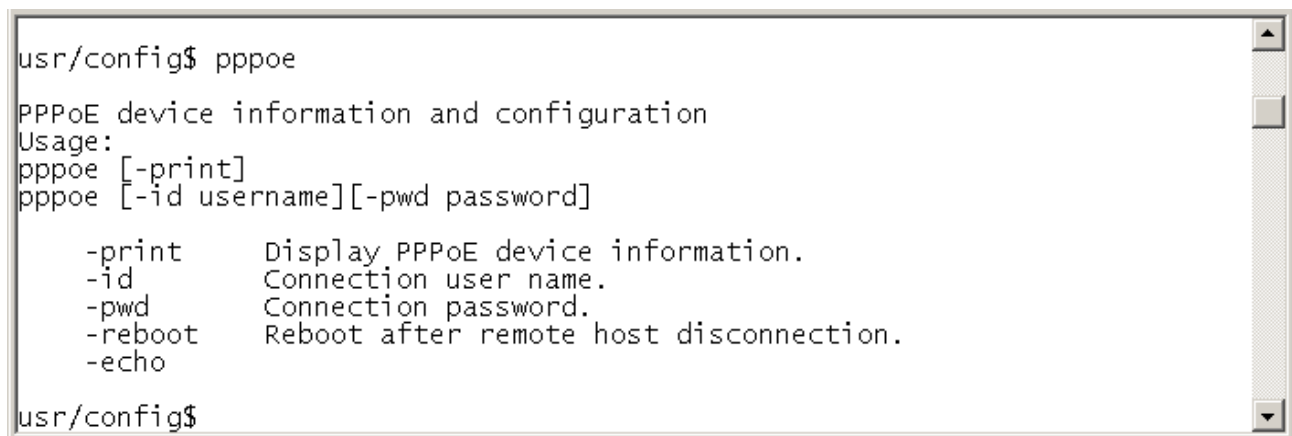
LAN information and configuration
Usage:
ifaddr [-print][[-dhcp used]][-sntp mode [server]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-dns index [dns server address]]
ifaddr [-autodns used]

    -print      Display LAN information and configuration.
    -ip         Specify ip address.
    -mask       Set Internet subnet mask.
    -gate       Specify default gateway ip address
    -ipmode     Set get IP mode(0:Fixed IP/1:DHCP/2:PPPoE)
    -sntp       Set SNTP server mode and specify IP address.
    -autodns    Specify the way to obtain DNS Server (0:Manual/1:Auto).
    -dns        specify IP address of DNS Server.
    -timezone   Set local timezone.
    -ipsharing  Specify usage of an IP sharing device and specify IP address
    .
    -upnp       Specify the upnp mode of ipsharing(0:Off/1:On)
    -server     Specify EMS Server IP address
    -id         Specify EMS Server ID
    -pwd        Specify EMS Server password
    -emstime    Specify EMS cycle time

Note:
Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
ifaddr -ipmode 1
ifaddr -sntp 1 210.59.163.254
ifaddr -ipsharing 1 210.59.163.254
ifaddr -upnp 1
ifaddr -autodns 1
ifaddr -dns 1 168.95.1.1
```


10. [pppoe] command

1. **-print:** display all current configurations and information.
2. **-id:** to set PPPoE authentication user name.
3. **-pwd:** to set PPPoE authentication password.
4. **-reboot:** Choose enable or disable this function. If user enables this function, after PPPoE disconnected, SIP-Phone will automatically reboot to re-connect, and after reboot, if SIP-Phone still can't connect with server, SIP-Phone will keep trying to connect. On the other hand, if user disables this function, SIP-Phone won't reboot and keep trying to connect. (**pppoe -reboot 0/1**)
5. **-echo:** Enable or Disable PPPoE Echo function. If user enabled this feature, SIP Phone will send out echo request to check PPPoE connection status. Please notice that if user disables this function, SIP Phone cannot detect if PPPoE connection is still alive or not. (**pppoe -echo 0/1**)



```
usr/config$ pppoe
PPPoE device information and configuration
Usage:
pppoe [-print]
pppoe [-id username][-pwd password]

    -print      Display PPPoE device information.
    -id         Connection user name.
    -pwd        Connection password.
    -reboot     Reboot after remote host disconnection.
    -echo

usr/config$
```

11. [flash] command

This command will clean the configuration stored in the flash rom to default value and reboot the SIP-Phone.

Note:

1. After user upgrade new software version, suggested to execute this command to make sure new software work well on SIP-Phone.
2. To execute the command **flash -clean**, all configuration of SIP-Phone stored in flash will be cleaned. It is authorized for the user whose login name is "root" only.


```
usr/config$ flash
Flash memory information and configuration
Usage:
flash -clean
Note:
  This command will clean the configuration stored in
  the flash and reboot it.
usr/config$
```

12. [sysconf] command

1. **-print:** display all current configurations.
2. **-idtime:** set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, SIP-Phone will dial out all number pressed.
3. **-service:** set SIP Phone to work as normal mode or hotline mode. If user wants to use hotline function, please set service to be 1, and set bureau related configuration. Please refer to **19. [bureau] command**.
4. **-keypad:** set DTMF type. User can select DTMF type SIP-Phone receive and transmit. (**sysconf -keypad 0/1**, 0 for in band, 1 for RFC2833, 2 for SIP-Info.)
5. **-2833type:** change RFC2833 Payload type.
6. **-eod:** select end of dialing key, e.g. set end of dial key as "*" button, after finished pressing dialing number then press "*" will dial out. (**sysconf -eod 0/1/2** , 0 for no end of dial key , 1 for "*" button, 2 for "#" button)
7. **-service:** set SIP Phone to be normal mode or under hotline mode. (**sysconf -service 0/1**, 0 for normal service, 1 for hotline service.)

Note:

To set SIP-Phone as hotline mode, user has to set SIP-Phone as Peer-to-Peer mode, and hotline table under **bureau** command.

```
usr/config$ sysconf

System information and configuration
Usage:
  sysconf [-idtime digit] [-keypad dtmf]
          [-2833type type] [-eod digit]
  sysconf -print

-print          Display system overall information and configuration.
-idtime         Inter-Digits time.(1~10 sec)
-service        Specify lanphone service type. (0: Normal service,
                1: HotLine service.)
-keypad         Select DTMF type: 0=In-band,
                1=RFC2833.
-2833type       RFC2833 Payload Type (range:96~128 inter-used:100,102~105)
-eod            End of Dial Digit setting(0: NONE, 1: *, 2: #)
-privacy        Privacy Number Type(0: NONE, 1: Japan)
Example:
  sysconf -keypad 0 -eod 2
usr/config$
```

13. [sip] command

1. **-print**: display all current configurations.
2. **-mode**: configure SIP-Phone as Proxy or Peer-to-Peer Mode.
Usage: **sip -mode 0/1**(1 for Proxy mode, 0 for Peer-to-Peer mode)
3. **-px**: set proxy server IP address or URL address (**sip -px "IP address or URL of Proxy server"**).
4. **-pxport**: set listening port of Proxy server.
5. **-outpx**: set IP address of outbound proxy server. After user set outbound proxy, all packets form SIP-Phone will be sent to outbound proxy server.
6. **-prefix**: set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, SIP-Phone will sent out messages as Account Name @ "IP address of Proxy", if user set prefix as abc, SIP-Phone will set out as abc123@ "IP address of Proxy". This function is for special proxy server.
7. **-line**: identify one number for the SIP-Phone to register to the Proxy (**SIP -line "line number"**).

Note: In proxy mode please remember to set user account information under security command.

8. **-pbsearch**: enable/disable phone book search function under Proxy Mode. If user enabled this function, SIP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched

data in phone book, SIP-Phone will make call to related IP address.

9. **–expire:** set expire time of registration. SIP-Phone will keep re-registering to proxy server before expire timed out.
10. **–port:** set listening UDP port or SIP-Phone.
11. **–rtp:** set RTP port number. SIP-Phone will use this port to send and receive voice.

```
usr/config$ sip
SIP stack information and configuration
Usage:
sip [-mode pxmode]
sip [-px address] [-prefix prefixstring]
  [-pxport ProxyPort] [-outpx address] [-line number]
  [-expire t1] [-port udpPort] [-rtp rtpPort]
sip -print
- - -
-print      Display SIP stack information and configuration.
-mode       Configure as Proxy mode or Peer-to-Peer mode.
-px         Proxy server address. (Proxy IPv4 address or Proxy dns name)
-pxport     Proxy server port. (the port of proxy)
-outpx      OutBound Proxy server address. (Proxy IPv4 address or Proxy dns name)
            specify as null)
-prefix     specify prefix string, use it when userID contains alphabets
            (if userID uses numerals, specify as null)
-line       TEL Phone number.
-pbsearch   Search phone book 0:off/1:on.
-expire     The relative time after which the message expires(0 ~ (2^31-1))
-port       SIP local UDP port number (5060~5070), Default: 5060
-rtp        RTP receive port number (2326~65534), Default: 16384
Example:
  sip -mode 1
  sip -px 210.59.163.171 -line 70
usr/config$
```

14. [security] command

1. **–print:** display all current configurations.
2. **-name:** set user ID of SIP-Phone for registering. User can set user name and password for registering. If password is no need, please set user name the same as line number or SIP-Phone won't register successfully.
3. **–pwd:** set account password for registering.

```
usr/config$ security
Security information and configuration
Usage:
security [-name username] [-password password]
security [-print]
- - -
-print      Display system account information and configuration.
-name       specify user name.
-pwd        specify password.
Example:
  security -name 1001 -pwd 1001
usr/config$
```

15. [voice] command

The voice command is associated with the voice codec setting information.

1. **-print**: display voice codec information and configuration.
2. **-send**: three voice packet size can be configured as 20 ms, 40 ms or 60 ms. (only 30 and 60 ms for G.723.1)
3. **-priority**: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need, for example, **voice -priority g723** or **voice -priority g723 711a g711u g729 g729a** means SIP-Phone can support only one codec up to five codecs.
4. **-volume**: There are three types can be adjustable, voice volume, ring volume, input gain and DTMF volume. Voice volume means the volume user can hear, input gain means the volume the other side can hear from SIP-Phone, DTMF means DTMF transmitting volume. (**voice -volume voice "value of volume"**, **voice -volume ring "value of volume"**, **voice -volume input "value of volume"**, **voice -volume dtmf "value of volume"**)

Note:

If value of volume set as 0 means -32db, 1 means -31db...etc.

5. **-nscng**: enable or disable sound compression and comfort noise generation. It is only for codec G.723.1. (0 for off, 1 for on)
6. **-echo**: enable or disable echo cancellation function .
7. **-mindelay**: set minimum delay of jitter buffer(0~150)
8. **-maxdealy**: set maximum delay of jitter buffer(0~150)

Note: It is for advanced administrator use only. Please ask your distributor before changing any settings of this command.

```

voice -print
voice -priority [G723] [G729] [G711U] [G711A] [G729A]

    -print      Display voice codec information and configuration.
    -send       Specify sending packet size.
                 G.723 (30/60/90 ms)
                 G.729 (20/40/60 ms)
                 G.711U (20/40/60 ms)
                 G.711A (20/40/60 ms)
                 G.729A (20/40/60 ms)
    -priority   Priority preference of installed codecs.
                 G.723
                 G.729
                 G.711U
                 G.711A
                 G.729A
    -volume     Specify the following levels:
                 voice volume (0-45, default: 32),
                 ring volume (0-45, default: 32),
                 input gain (26-45, default: 32),
                 dtmf volume (0-31, default: 31),
    -nscng      No sound compression and CMG. (G.723.1 only, on=1, off=0).
    -echo       Setting of echo canceller. (on=1, off=0, per port basis).
    -mindelay   Setting of jitter buffer min delay. (0-150, default: 90).
    -maxdelay   Setting of jitter buffer max delay. (0-150, default: 150).

Example:
voice -send g723 60 g729 60 g711u 60 g711a 60 g729a 60
voice -volume voice 20 input 32 dtmf 27
voice -echo 1
usr/config$ voice -volume ring 20
usr/config$ commit

This may take a few seconds, please wait....
commit to flash memory ok!
usr/config$

```

16. [tos] command

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority. As per the definition in RFC2474, the DS field is Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined in RFC2597 to classify the traffic into different service classes. The mapping of Code Point value of DS-field to egress traffic priorities is shown as follows.

1. High priority with DS-field.

Expected Forwarding (EF)	101110	====>	46 (Decimal System)
Assured Forwarding (AF)	001010	====>	10 (Decimal System)
	010010	====>	18 (Decimal System)
	011010	====>	26 (Decimal System)
	100010	====>	34 (Decimal System)

2. Low Priority with DS-field:

Assured Forwarding (AF)	001100	====>	12 (Decimal System)
	010100	====>	20 (Decimal System)
	011100	====>	28 (Decimal System)
	100100	====>	36 (Decimal System)
	001110	====>	14 (Decimal System)

010110 ==> 22 (Decimal System)
 011110 ==> 30 (Decimal System)
 100110 ==> 38 (Decimal System)
 000000 ==> 0 (Decimal System)

DROP Precedence	Class #1	Class #2	Class #3	Class #4
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

1. **-print** : display all current configurations.
2. **-rtptype**: set DSCP value of signaling packets from 0 to 63
3. **-sigtype**: set DSCP value of RTP packets from 0 to 63

Note:

1. This command won't be functional until network environment can be capable with DSCP function.
2. tos -rtptype 14 -sigtype 10 is top priority of package.

```
usr/config$ tos
IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
tos [-rtpreliab mode]
tos -print
Example:
tos -rtptype 10 -sigtype 0
usr/config$ █
```

17. [ddns] command

This function is for Dynamic Domain Name Server service. Once user register to one DDNS server, he can specify domain name for the SIP Phone. When SIP Phone reboot, it will automatically update it's IP address to DDNS server. In this way, even SIP Phone is using dynamic IP address, other endpoint can locate this SIP Phone by its domain name.

1. **-print**: display DDNS overall information and configuration.
2. **-enable**: to enable/disable DDNS function. (**ddns -enable 0/1**, 0 for disable and 1 for enable).
3. **-server**: to set IP address of DDNS login server. (Now only one DDNS server is available---www.dyndns.org)
4. **-hostname**: to set the registered Domain Name of SIP Phone. (Ex. **ddns -lp001.ddns.org**)
5. **-id**: to set login ID of registered account to log in DDNS server.
6. **-passwd**: to set password of registered account to log in DDNS server.
7. **-checkip**: to enable/disable check IP function. If SIP Phone is behind IP sharing, when this function is enabled, SIP Phone will check it's public IP address by asking IP address check server and send to DDNS server to update DDNS data. If this function is disabled, when SIP Phone is behind IP sharing, it will send it's private IP address to DDNS server.
8. **-checkipsrv1/2**: to set IP address of primary and secondary IP address check server.
9. **-delay**: to set the update interval time. SIP Phone will re-update its IP address in this time. (**ddns -delay 1-59m/1-24h**, m means minute, h means hour)
10. **-force**: to force to execute DDNS update. Once user enters this command, SIP Phone will update DDNS data immediately. (**ddns -force "IP address of SIP Phone"**)

Note:

1. For now we only support DDNS server as www.dyndns.org.
2. User must register to DDNS server first, and specify user name and password in **ddns -id** and **ddns -passwd**.
3. The default IP address of DDNS login server is member.dyndns.org.
4. User has to specify domain name applied for SIP Phone in **ddns -hostname**.
5. The default IP address of check IP server in is checkip.dyndns.org.

18. [vlan] command

1. **-print**: display current VLAN configuration settings.
2. **-dev**: enable/disable VLAN configuration.
3. **-vid**: set CPU VLAN ID number.
4. **-pcvid**: set PC port VLAN ID number.

Note: Only when vid numbers are the same, network can connect. For example, if PC A connect to PC port of SIP Phone A, and vid of pcvid are different, this PC can't reach this SIP Phone, however, if PC B connect to SIP Phone B, and pcvid of SIP Phone B is the same with vid of SIP Phone A, PC B can reach SIP Phone A.



5. **-priority**: set CPU packets priority number.(From 0 to 7, 7 is the highest priority.)
6. **-pcpriority** : set pc port priority number. (From 0 to 7, 7 is the highest priority.)
7. **-pcdroptag**: enable/disable pc port drop tag function. If this function is enabled, SIP Phone will drop priority tag on packets sending out from PC port.


```

usr/config$ vlan
VLAN information and configuration
Usage:
vlan [-dev enable] [-vid number] [-pcvid number]
      [-priority number] [-pcpriority number] [-pcdroptag enable]
      -print          Display VLAN information and configuration.
      -dev            Enable/disable VLAN configuration. ( 0:Disable / 1:Enable)
Example:
      -vid            VID number.(1~4095, default:1)
      -pcvid          PC port VID number.(1~4095, default:1)
      -priority       Priority number.(0~3:Low priority, 4~7:High priority)
      -pcpriority     PC port priority number.(0~3:Low priority, 4~7:High priority)
      -pcdroptag      Drop PC port tag.(0:Disable / 1:Enable)
Example:
      vlan -dev 1
      vlan -vid 2
      vlan -pcvid 2
      vlan -priority 4
      vlan -pcpriority 0
      vlan -pcdroptag 1
usr/config$

```

19. [bureau] command

1. **-print**: show all current configurations.
2. **-hotline**: set hotline table. User can set hotline function to specify one IP address for SIP-Phone to dial out directly. Once user picks up SIP-Phone, it will automatically dial out to the assigned IP address and number.
(bureau –hotline “IP address” “phone number of remote site”)

Note:

To set SIP-Phone as hotline mode, user has to set SIP-Phone as Peer-to-Peer mode, and **sysconf –service 2** command.

```

usr/config$ bureau
Bureau line setting information and configuration
Usage:
bureau [-hotline [Port DestIP TELnum]]
bureau -print
      -print          Display Bureau line information and configuration.
      -hotline        Set Hot line information. (Port range: 1~8)
Note:
      Hotline feature should be used together with:
          $sysconf -service 2 (HotLine service)
          $sip -mode 1 (peer-to-peer mode)
Example:
      bureau -hotline 1 192.168.4.69 628
usr/config$

```

20. [rom] command

1. **-print**: show all current configurations and version information.
2. **-app, -boot, -dsptest, -dspcore, -dspapp**,: upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone .

Note:

After upgrade Application, please remember to execute flash –clean command, which will clean all configurations become factory values except IP address.

3. **-boot2m**: to upgrade 2mb rom file, which includes all firmware file mentioned in item 2.
 4. **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.
 5. **-f**: the file name prepared for upgrading is necessary as well.
 6. **-method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
 7. **-ftp**: specify user name and password for FTP download method.
- For example: User prepares to upgrade the latest app rom file – sip1p.100, the TFTP server is 192.168.4.1.1.

rom –app –s 192.168.1.1 –f sip1p.100



```
usr/config$ rom
ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp]
-s TFTP/FTP server ip -f filename
rom -print
    -print      show versions of rom files. (optional)
    -app       update main application code(optional)
    -boot      update main boot code(optional)
    -boot2m    update 2M code(optional)
    -dsptest   update DSP testing code(optional)
    -dspcore   update DSP kernel code(optional)
    -dspapp    update DSP application code(optional)
    -s         IP address of TFTP/FTP server (mandatory)
    -f         file name(mandatory)
    -method    download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
    -ftp       specify username and password for FTP
Note:
This command can run select one option in 'app', 'boot',
, 'dsptest', 'dspcore', and 'dspapp'.
Example:
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin
usr/config$
```

Command **rom –print** can show current version installed in SIP-Phone .

```
usr/config$ rom -print

download Method : TFTP
  Boot Rom : lp201_boot.101
Application Rom : lp201_sip0202.BIN
  DSP App : 48302ce3.140
  DSP Kernel : 48302ck.140
  DSP Test Code : 483cbit.bin
usr/config$
```

21. [passwd] command

For security protection, user has to input the password before entering **application user/config mode**. Two configurations of login name/password are supported by the system.

1. **-set**: set password of “root” users or “user” users. (**passwd -set root/user “password”**)
2. **-clean**: clean up password restored before, and user can login :”root/user, password: ”press enter”.

User who requests authorization to execute **all** configuration commands needs to login with “root”. If a user login with “user”, only commands below are functional:

1. ifaddr
2. pbook
3. password set user “password”
4. commit
5. reboot

```
usr/config$ passwd

Password setting information and configuration
Usage:
  passwd -set Loginname Password
  passwd -clean
Note:
  1. Loginname can be only 'root' or 'administrator'
  2. passwd -clean will clear all passwd stored in flash,
     please use it with care.
Example:
  passwd -set root Your_Passwd_Setting
usr/config$
```

Chapter 5 Upgrading Software on the SIP-Phone

SIP-Phone supports three methods to upgrade the new version. All methods are necessary to prepare the **TFTP** or **FTP** program on the host PC as **TFTP/FTP server**. After installing **TFTP/FTP** program on one PC and connecting to network, SIP-Phone is ready to be upgraded.

1. LCD Panel Control
2. Remote Control: Telnet
3. Web Management

1.From LCD Phone Menu

Please choose the 6 2 selection-**Firmware Upgrade**. Press **OK** to enter into the sub-selection as below.

1. Firmware Update

(1) Download method

There are two methods to download new version file, please move the ">" symbol by press ← or → on the keypad to select TFTP or FTP method, then press **OK** to confirm it.

(2) FTP/TFTP Sever

User has to offer one TFTP/FTP server IP Address and set this IP Address via keypad. The IP Address is necessary for upgrading SIP-Phone new application rom file.

(3) FTP Account

User has to input user name for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

(4) FTP Password

User has to input password for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

(5) File Name

User has to press the file name of new application rom file prepared for upgrading

(6) Application Version

Show versions of application software.

Note:

Download via LCD command can only upgrade new **application** rom file.

(7) Start to Upgrade

Select YES or NO to start upgrade. After download is finished, press OK then SIP-Phone will ask if need to reboot.

2. Web Management

Please refer to chapter 4.18 [rom] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left sidebar and the 'ROM Configuration' section on the right. The sidebar menu includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration (highlighted), Flash Clean, Commit Data, and Reboot System. The 'ROM Configuration' section contains the following fields:

ROM Configuration	
FTP/TFTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Target File name:	<input type="text"/>
Method:	<input type="text" value="TFTP"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	<input type="text" value="Application Image"/>
<input type="button" value="OK"/>	

- **FTP/TFTP Server IP Address:** Set TFTP server IP address
- **Target File name:** Set file name prepared to upgrade
- **Method:** Select download method as TFTP or FTP
- **FTP Login:** Set FTP login name and password
- **Target File Type:** Select which sector of SIP-Phone to upgrade

Note:

1. After 2mb file download is finished, all configurations might change to default values, user has to configure again.
2. After upgrade Application, please remember to execute Flash Clean,

which will clean all configurations become factory values except IP address.

3. Remote Control: Telnet

Please refer to chapter 4.18 **[rom] command**

1. **-print**: show all current configurations and version information.
2. **-app, -boot, -dsp test, -dsp core, -dsp app**,: upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone .

Note:

After upgrade Application, please remember to execute flash –clean command, which will clean all configurations become factory values except IP address.

3. **-boot2m**: to upgrade 2mb rom file, which includes all firmware file mentioned in item 2.
4. **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.
5. **-f**: the file name prepared for upgrading is necessary as well.
6. **–method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
7. **–ftp**: specify user name and password for FTP download method.
For example: User prepares to upgrade the latest app rom file – sip1p.100, the TFTP server is 192.168.4.1.1.
rom –app –s 192.168.1.1 –f sip1p.100

```
usr/config$ rom
ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp]
-s TFTP/FTP server ip -f Filename
rom -print
  -print      show versions of rom files. (optional)
  -app        update main application code(optional)
  -boot       update main boot code(optional)
  -boot2m     update 2M code(optional)
  -dsptest    update DSP testing code(optional)
  -dspcore    update DSP kernel code(optional)
  -dspapp     update DSP application code(optional)
  -s          IP address of TFTP/FTP server (mandatory)
  -f          file name(mandatory)
  -method     download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
  -ftp        specify username and password for FTP
Note:
This command can run select one option in 'app', 'boot',
, 'dsptest', 'dspcore', and 'dspapp'.
Example:
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin
usr/config$
```

Command **rom -print** can show current version installed in SIP-Phone .

```
usr/config$ rom -print

Download Method : TFTP
  Boot Rom      : lp201_boot.101
Application Rom : lp201_sip0202.BIN
  DSP App       : 48302ce3.140
  DSP Kernel    : 48302ck.140
  DSP Test Code : 483cbit.bin
usr/config$
```